1   METHOD OF OPTIMIZING COMPRESSION RATE IN ADAPTIVE

2   DIFFERENTIAL PULSE CODE MODULATION (ADPCM)

3   BACKGROUND OF THE INVENTION

4   1. Field of the Invention

5   The present invention relates to a method of optimizing the compression

6   rate in Adaptive Differential Pulse Code Modulation (ADPCM), especially to a

7   modified pulse code modulation technique that employs a prognostic code

8   converter to generate variable length codes on top of the ADPCM coding, based

9   on the probability of occurrence of data bits, so as to further reduce the

10   compressed data size and to increase the compression rate of the conventional

11   ADPCM coding.

12   2. Description of Related Arts

13   The main difference between the pulse code modulation (PCM) and the

14   Adaptive Differential Pulse Code Modulation (ADPCM) lies in the compression

15   format, where the PCM coding is mainly used to convert analog signals to digital

16   signals in the audio signal processing using a fixed sampling frequency. This

17   PCM coding has the advantage of realistic sound reproduction, but a large

18   memory is required for such operation. ADPCM coding is an adaptive coding

19   technique that incorporates an adaptive predictor in the compression coding

20   process to produce compressed data with suitable compression rate. For example,

21   a 16-bit data sample can be compressed according to a 4:1 ratio to a 4-bit data

22   sample; and conversely, the compressed data can be decompressed according to a

23   1:4 ratio to restore it to the original signal waveform of a 16-bit string.

24   The application of ADPCM for compressing audio and speech signals is

1 aimed at easing the transmission bandwidth and reducing the memory storage

2 requirements. However, the main disadvantage of ADPCM lies in the low

3 compression rate. Although the memory requirements for the ADPCM codes are

4 fewer than those for the PCM codes, the compression rate is still not sufficient to

5 meet the memory requirements for portable electronic devices, for example CD

6 players.

7 From numerous observations, it is found that data contents in adjacent

8 data blocks are subject to less variation than non-adjacent data blocks. If the

9 locality of data in the compression coding can be analyzed, the compression rate

10 can be further improved, A modified pulse code modulation technique based on

11 Huffman coding can be employed by first creating a look-up table with variable

12 length codes that match the occurrence rate of data bits. This modified pulse code

13 modulation process, using variable-length or run-length coding, can be a

14 second-level coding process on top of the conventional ADPCM for achieving a

15 higher compression rate in the audio signal processing.

16 In general, both coding techniques have been used in digital signal

17 processing. For example, a portable CD player may be installed with an

18 electronic anti-shock system (EASS) to maintain constant quality of audio output

19 and to create a buffering effect against physical vibrations of the CD player. The

20 electronic anti-shock system (EASS) shown in Fig. 4 with a single output

21 channel has an ADPCM encoder (71) on the front end and an ADPCM decoder

22 (73) on the back-end. The ADPCM encoder (71) is used to generate ADPCM

23 codes from input data samples which are then saved in the random access

24 memory (72); and conversely, the compressed data retrieved from the random

1   access memory (72) are processed by the ADPCM decoder (73) for decoding the

2   ADPCM codes to the original waveform of audio signals en route to an audio

3   signal processor (74) for audio output.

4   The compression coding for the above-mentioned ADPCM encoder (71)

5   can be both a 4-bit mode and 3-bit mode. Since all current audio equipment has at

6   least two audio channels, the 4-bit operation mode is selected in this example for

7   calculation of the bit rate with a sampling frequency of 44.1 KHz:

8   4(bits) × 44100 × 2 (number of audio channels) =352,800 Kbps

9   If the operation is in 3-bit operation mode, the bit rate is:

10   3(bits) × 44100 × 2 (number of audio channels) =264,600 Kbps

11   If the memory capacity in the above anti-shock buffering system is 16M,

12   Bits, as in the present example, the required buffering time for the two operation

13   modes can be:

14   4-bit mode: 16,000,000 ÷ 352,800 = 45.35 (sec)

15   3-bit mode: 16,000,000 ÷ 264,600 = 60.46 (sec)

16   From the foregoing, it is apparent that the bit rate and the memory

17   capacity directly affects the buffering time of the EASS. Since the data saving

18   operation in the EASS has to be sustained for certain time period in order to

19   create an anti-shock buffering effect, and the buffering time is dependent on the

20   compression rate in the coding process and the capacity of random access

21   memory used, the performance of an ADPCM-based CD player embedded with

22   EASS can also be affected.

23   However, the main disadvantage of ADPCM lies in the low compression

24   rate, as the resulting compression rate of the conventional ADPCM is not enough

1   for efficient utilization of random access memory as in the case of the

2   ADPCM-based CD players. There are some more advanced compression

3   algorithms, such as the MPEG/layer I and II, which are able to produce

4   reasonably good sound quality using much less memory than ADPCM.

5   Under the precondition not to increase random access memory, the only

6   way to improve the performance of EASS for ADPCM-based CD players,

7   without incurring additional hardware costs, is to increase the bit rate in the

8   compression process. If the compression rate of ADPCM is increased, the

9   performance of EASS can be improved with lossless code modulation. Another

10 benefit of EASS is the power saving consideration for portable CD players as the

11 longer the CD servo can be suspended by the buffering effect, the less system

12 power is used.

13 SUMMARY OF THE INVENTION

14   The main object of the present invention is to provide a method of

15 optimizing the compression rate in Adaptive Differential Pulse Code Modulation

16 (ADPCM) that uses variable length coding based on the probability of

17 occurrence of data bits to further reduce the size of compressed data and improve

18 the performance over conventional adaptive differential pulse code modulation

19 (ADPCM).

20   To this end, the modified pulse code modulation technique employs a

21 prognostic code converter to generate variable length codes on top of the

22 adaptive differential pulse code modulation (ADPCM) coding. Variable length

23 codes are generated according to the probability of occurrence of the data bits. In

24 the variable length coding process, a bit string with the highest occurrence rate

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1   will be assigned a variable length code with the shortest length, whereas the bit

2   string with the lowest occurrence rate will be assigned variable length code with

3   the longest length. Conversely, when decoding, the compressed data is first

4   processed by a variable length decoder to reconstruct the pure ADPCM coding

5   and then to the original audio signal in a lossless audio signal processing.

6   The features and structure of the present invention will be more clearly

7   understood when taken in conjunction with the accompanying figures.

8   <u>BRIEF DESCRIPTION OF THE DRAWINGS</u>

9   Fig. 1 is system architecture of the present invention;

10   Fig. 2 is an example of the probability table of the occurrence of bit

11   strings in a certain data block;

12   Fig. 3 is a table of the variable length codes matching the probability of

13   occurrence of data bits; and

14   Fig. 4 is a block diagram of the conventional electronic anti-shock

15   system (EASS).

16   <u>DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT</u>

17   The present invention provides a method of optimizing the compression

18   rate in a adaptive differential pulse code modulation (ADPCM) by employing a

19   prognostic code converter with a system architecture as shown in Fig. 1,

20   comprising:

21   an ADPCM encoder (11) for encoding with an ADPCM coding

22   technique;

23   a random access memory (12) for temporary saving of output codes from

24   the ADPCM encoder (11) to provide a buffering effect;

1   an ADPCM decoder (13) for restoring the compressed ADPCM codes

2   retrieved from the random access memory (12) to the original audio signals;

3   a variable-length encoder (14), which is installed in between the

4   ADPCM encoder (11) and the random access memory (12), for modulating the

5   variable length codes on top of the ADPCM coding, in accordance with the

6   probability of occurrence of data bits en route to the random access memory (12);

7   and

8   a variable-length decoder (15), which is installed in between the random

9   access memory (12) and ADPCM decoder (13), for decoding the variable length

10   codes retrieved from the random access memory (12) to the original ADPCM

11   codes, and then passing on to the ADPCM decoder (13) for restoring to the

12   original audio signals.

13   Applying this model to the example of an electronic anti-shock system

14   (EASS) embedded in ADPCM-based CD players, the variable length encoder

15   (14) is installed in between the ADPCM encoder (11) and the random access

16   memory (12) to further compress ADPCM codes before being saved in the

17   memory, whereby the data storage capacity of the memory can be increased and

18   the buffering time can be extended. Conversely, when the compressed data are

19   retrieved from the random access memory (12), the compressed data are first

20   converted back to ADPCM codes and then to the original audio signals for

21   lossless audio performance.

22   The detailed operation of the variable length encoder (11) and decoder

23   (13) are to be explained in the following paragraphs.

24   The variable length encoder is based on the Huffman coding scheme,

1 whereby pulse codes with the same length bit strings can be converted to variable

2 length codes according to the probability of occurrence of data bits. A variable

3 length code is designed to correspond to the original ADPCM code, but the size

4 of the compressed code is substantially less than a pure ADPCM code. In the

5 Huffman coding scheme, the bit string with the highest occurrence rate will be

6 assigned the variable length code with the shortest length, whereas the bit string

7 with the lowest occurrence rate be assigned the variable length code with the

8 longest length. An example of ADPCM coding is given below using the 3-bit

9 ADPCM mode.

10 In the 3-bit coding mode, the patterns of bit strings are defined as:

11 001 010 011 100 101 110 111

12 The probability of occurrence in a data block are in an ascending order as

13 follows 111, 001, 110, 010, 101, 011, and 100, where the occurrence of 111 has

14 a 0.35 probability; 001 and 110 have a 0.3 probability; 010 and 101 have a 0.2

15 probability; 011 and 100 has a 0.15 probability. Basing on the list of probability

16 of occurrence, the bit string with the highest occurrence rate is assigned a

17 variable length code represented by one data bit "0" to replace the original

18 ADPCM code; the bit string with the second highest occurrence rate is then

19 assigned a variable length code formed by three data bits "10S" (where S being

20 the most significant bit); and the third and fourth highest occurrence rates are

21 respectively assigned three bits codes of 110S and 111S respectively. The

22 conversion table for all variable length codes is shown in Fig. 3. To demonstrate

23 the effect of such a conversion process, a block of data is selected for comparing

24 compressed data before and after the variable length coding process.

1   Assuming the original ADPCM codes are represented by:

2   011_010_001_111_001_110_111_111_001_110_001_110_111

3   The above data block occupies 39 bits, and then the variable length codes

4   are generated on top of the ADPCM coding in accordance with the present

5   invention, the contents of the data block being:

6   1110_1100_100_0_100_101_0_0_100_101_100_101_0

7   and the output data becomes

8   111011001000100101001001011001010, totaling 33 bits on top of the

9   ADPCM coding process, reduced by 6 bits as compared with conventional

10   ADPCM, that means the compression rate is increased by 15%, (39-33)/39=0.15.

11   Applying the variable length coding to the above example of EASS,

12   with the increased compression rate, once the data storage capacity of the random

13   access memory is enhanced, the buffering time can be extended. When the

14   compressed data are retrieved from the random access memory, the compressed

15   data are first decoded to the original ADPCM codes and then to the original audio

16   signals for lossless audio output.

17   From the foregoing, the present invention mainly relies on the prognostic

18   code converter to generate variable length codes according to the probability of

19   occurrence of data bits, such that the bit string with the highest probability of

20   occurrence will become the variable length code with the shortest length, and the

21   bit string with lesser probability of occurrence will become the variable length

22   code with longer length. Using this coding technique, the average number of bits

23   in a data block can be substantially reduced and the compression rate can be

24   enhanced accordingly. In the case of EASS, the data storage capacity of random

1   access memory can be increased for extending the buffering time. Therefore, the

2   output performance of the CD player can be improved.

3   The foregoing description of the preferred embodiments of the present

4   invention is intended to be illustrative only and, under no circumstances, should

5   the scope of the present invention be so restricted.